Each of claims 1-2, 18-19, and 24-25 recites numerous limitations relating to the decoding and/or encoding of audio data based on a "superframe."[1] From the claims it can be seen that the recited superframe is itself composed of a plurality of individual frames. As an initial matter, there is no indication in the Office action that *Hardwick* teaches these central limitation relating to a superframe, and Applicants' independent review has verified that *Hardwick* contains no such teaching. Accordingly, in addition to the accepted fact that the limitation of a "superframe buffer" does not appear in *Hardwick, Hardwick* clearly also does not expressly teach any of the other limitations that relate to a superframe.

What *Hardwick* does teach is a system that quantizes voice data frames on a sub-frame basis. Thus the remaining issue is whether a teaching of quantizing voice data frames on a ***sub-frame*** basis expressly or inherently amounts to a teaching to treat voice data frames on a ***superframe*** basis (and/or the inverse for the decoding and combination encoding/decoding claims). As will be discussed below, the technique of *Hardwick* and the technique of the present application are more dissimilar than similar. In fact, they take opposite approaches, with *Hardwick* seeking economy by going from the frame to a smaller entity (the subframe), and the present application and claims encompassing a new approach, that entails going from the frame to a ***larger*** entity (the superframe).

[1] *See, for example,* claim 1 (...a **superframe** encoder ... wherein parametric voice data ... from the [**frame**-based] analysis module is selectively quantized to produce voice data ...);
claim 2 (...a **superframe** encoder for receiving voice data parameters from the [**frame**-based] analysis module ... for quantizing and encoding said data ...);
claim 18 (...a **superframe** encoder for receiving unquantized voice data parameters for groups of frames within a superframe, said superframe encoder comprising ... a parameter quantizer and encoder for quantizing and encoding voicing parameters received from said [frame-based] analysis module...);
claim 19 (...a superframe decoder for receiving ...a series of superframes and decoding and inverse quantizing said superframes into quantized frame-based voice parameters...);
claim 24 (...loading **multiple** frames of digitized voice into a superframe buffer...encoding digitized voice within each frame of the superframe buffer by parametric analysis to produce **frame-based** parametric voice data...quantizing the [parametric voice data, as well as other data,] into discrete values represented by a reduced set of data bits that form quantized superframe parameter data...); and
claim 25 (... receiving superframe-based parametric voice data...decoding and inverse quantizing the voice data... to recreate a set of frame-based voice parameter values...).

In order to fully understand the differences between *Hardwick's* sub-frame processing technique and the superframe methods and mechanisms of claims 1-2, 18-19, and 24-25, a brief discussion of the relevant aspects of *Hardwick* is warranted. The technique of *Hardwick* begins by breaking a frame of audio data down into a number of subframes. *See Hardwick,* Abstract, first three lines ("Speech is encoded into a frame of bits. A speech signal is digitized into a sequence of digital speech samples that are then divided into a sequence of subframes.") Next, *Hardwick* estimates model parameters for each subframe. *Id,* lines 3-4. Finally, the model parameters for the subframes are jointly quantized. *Id.,* lines 8-9.

In discussing his subframe-based approach, *Hardwick* makes it absolutely clear that the per-frame bit savings yielded by his technique only apply when the frame is broken down into a ***smaller*** entity, i.e. a subframe. At column 9, lines 34-49, *Hardwick* describes the efficiency-causing mechanism in greater detail:

> Empirical data indicates that the ***fundamental frequency does not vary significantly from subframe to subframe*** ...The vector quantizer can more accurately map these ***small changes*** in fundamental frequency ***between subframes***, since there is a higher density of quantization levels for small changes in fundamental frequency. Therefore, the vector quantizer reduces the number of bits required to encode the fundamental frequency without significant degradation in speech quality.

(emphasis added). It can be seen from the above quote that *Hardwick* does not teach the use of a larger entity, such as a superframe, and in fact ***requires*** the use of the smaller entity, or subframe, to achieve the stated efficiencies.

Accordingly, in addition to the admitted fact that *Hardwick* does not teach the use of a superframe buffer, *Hardwick* also fails to teach any aspect of the existence or use of superframes at all. Moreover, as discussed above, *Hardwick* actually requires the use of a subframe, rather than a superframe, as set forth in the subject claims. In addition, *Ojala* also does not supply the missing limitations, since it does not contain any discussion of the existence or use of superframes. Thus, since the limitations of claims 1-2, 18-19, and 24-25 for which

Hardwick was cited are not taught in *Hardwick*, nor in *Ojala*, it is respectfully requested that the rejections of these claims be reconsidered and withdrawn.

Dependent Claims 3-17, and 21

Each of claims 3-17, and 21 depends from one of the claims discussed in the prior section, and as such incorporates the limitations of the respective base claim. Each base claim contains limitations that are not taught by *Hardwick*, which was cited to supply the relevant limitations, or by *Ojala*, and it is thus respectfully submitted that claims 3-17 and 21 are patentable for the reasons discussed above with respect to claims 1-2, 18-19, and 24-25. Accordingly, it is respectfully requested that the rejections of claims 3-17, and 21 be reconsidered and withdrawn.

Moreover, with respect to claims 7-9, it is respectfully submitted that (a) the reference cited to supply the additional limitations of these claims, i.e. *Ojala*, does not contain the necessary teachings, and (b) the cited teaching to combine *Ojala* with *Hardwick* is not legally sufficient. With respect to the first point, each of claims 7-9 recites much more than simply the existence or use of an LPC technique. There is no allegation in the Office action that either reference supplies these missing limitations, and Applicants have been unable to precisely identify all of the necessary teachings in their own review of the references.

With respect to the teaching to combine, the Office action states that "it would have been obvious … to implement linear prediction quantization as suggested by Ojala et al, for the purpose of implementation in variable bit-rate wireless telephone networks … as also taught by Ojala et al." It is respectfully submitted that this cited teaching to combine does not pertain to any actual combination, but rather simply states alleged benefits and attributes of *Ojala* without reference to *Hardwick*, and without stating or showing why any ***combination*** would be obvious, if and how the combination would actually ***work***, and why one of skill in the art would have any

expectation of ***success*** in making the combination. In fact, different voice coding systems can have quite different parameters and functionalities, and it is not readily apparent from the Office action, or from *Hardwick* or *Ojala* how or in what way the coding model of *Ojala* should be modified to fit the teachings of *Hardwick*, or how the efficiency mechanisms of *Hardwick* should be adapted to operate on the model parameters of *Ojala*.

For these additional reasons, it is respectfully submitted that claims 7-9 are not obvious in view of *Hardwick* or *Ojala*, or any combination thereof. Accordingly, it is requested that the rejections of these claims be reconsidered and withdrawn.

Independent Claim 20 and Dependent Claim 21

Independent claim 20 is generally similar to the other independent claims, but pertains more specifically to a process for decoding a parametric voice encoded data stream. For ease of reference, please refer to claim 20 as reproduced in amended form above or in the Appendix. Claim 20 sets forth a very specific and novel set of steps for efficiently decoding parametric voice data, the process being grounded on three primary steps, namely:

- buffering received parametric voice data, wherein the voice data has a plurality of pitch periods,
- creating, in a specified manner, an estimated spectrum of excitation within each pitch period by breaking the frequency spectrum into regions based on a cutoff frequency, and deriving a time domain representation, and
- generating an analog voice signal from the time domain representation.

The Office action does not identify this sequence, or even all of its constituent steps, anywhere in the cited references, nor have Applicants identified such. Thus independent claim 20 recites limitations for which no teaching appears in either reference or in their cited combination. Accordingly, it is respectfully submitted that claim 20 is patentable over the

cited references, and it is requested that the rejection of claim 20 be reconsidered and withdrawn.

With respect to claim 21, this claim depends from claim 20, and is believed to be patentable for at least the same reasons as the base claim. Accordingly, it is respectfully requested that the rejection of claim 20 be reconsidered and withdrawn.

Independent Claims 22 and 23

It is respectfully noted that the recited limitations of claims 22 and 23 are not alleged to be found in the cited references, nor has Applicants' independent review revealed any such teachings. Claim 22 pertains to an up-transcoder apparatus for converting a superframe-encoded voice data stream into a frame-based encoded voice data stream. Similarly, claim 23 pertains to a down-transcoder apparatus for converting a frame-based encoded voice data stream into a superframe-encoded voice data stream.[2] There are no such conversion apparatus taught in the cited references. In fact, it appears doubtful that the encoded data streams discussed in *Hardwick* are even transcod*able*, since, as discussed above, they are based on subframes, which are not typically *independently* codable entities. Moreover, claims 22 and 23 recite limitations pertaining to the creation or processing of superframes, which limitations are not taught or suggested by any cited reference, as discussed more generally above with respect to claims 1-2, 18-19, and 24-25.

In any case, because the limitations of claims 22 and 23 are not taught or suggested in the cited references, taken either singly or in combination, it is respectfully requested that the rejections of these claims be reconsidered and withdrawn.

[2] Transcoding entails creating a coded signal at a first bit rate, and/or in a first format, from a coded signal at a second bit rate, and/or in a second format, without creating and re-coding an intermediate raw data set.

Independent Claims 26-30

Independent claims 26-30 add no new matter and are believed to be patentable for at least the reasons discussed above with respect to independent claims 1-2, 18-19, and 24-25.

CONCLUSION

The application is considered in good and proper form for allowance, and the examiner is respectfully requested to pass this application to issue. If, in the opinion of the examiner, a telephone conference would expedite the prosecution of the subject application, the examiner is invited to call the undersigned attorney.

Respectfully submitted,

Phillip M. Pippenger, Reg. No. 46,055
One of the Attorneys for Applicants
LEYDIG, VOIT & MAYER, LTD.
Two Prudential Plaza, Suite 4900
180 North Stetson
Chicago, Illinois 60601-6780
(312) 616-5600 (telephone)
(312) 616-5700 (facsimile)

Date: 8/1/02

APPENDIX
(Marked copy of claims 3, 7, 12, 18, 20-21, and 23 as amended by way of the present amendment)

3. (Once Amended) A voice compression apparatus as recited in claim 2, wherein the analysis module [is capable of receiving voice data parameters] is selected from the group of voice encoders consisting of linear predictive coders, mixed-excitation linear prediction coders, harmonic coders, and multiband excitation coders.

7. (Once Amended) A voice compression apparatus as recited in claim 2, wherein said super-frame encoder includes a quantizer of linear prediction parameters, wherein quantization is performed with a codebook-based interpolation of linear prediction parameters that employ different interpolation coefficients for each linear prediction parameter, and wherein said quantizer operates in closed loop mode to minimize overall error over a number of frames.

12. (Once Amended) A voice compression apparatus as recited in claim 11, wherein said pitch smoother classifies frames into onset and offset frames based on at least four waveform feature parameters selected from the group of waveform feature parameters consisting of energy[.] zerocrossing rate, peakiness, maximum correlation coefficient of input speech, maximum correlation coefficient of 500 Hz low pass filtered speech, energy of low pass filtered speech, and energy of high pass filtered speech.

18. (Once Amended) A voice compression apparatus, comprising:

(a) a superframe buffer for receiving multiple frames of voice data;

(b) a frame-based analysis module for determining a set of voice data parameters for said voice data; and

(c) a super-frame encoder for receiving unquantized voice data parameters for groups of frames within a [superframes] superframe, said superframe encoder comprising

(i) a pitch smoother for determining pitch and U/V decisions for each frame of the superframe and for extracting [extracts] parameters needed for frame classification into onset and offset frames,

(ii) a bandpass voicing smoother for determining bandpass voicing strengths for the frames within the superframe and for determining [determines] cutoff frequencies for each frame, and

(iii) a parameter quantizer and encoder for quantizing and encoding voicing parameters received from said analysis module, said pitch smoother, and said bandpass voicing smoother into a set of bits and encoding said bits into an outgoing digital bit stream for transmission.

20. (Once Amended) A method of decoding a parametric voice encoded data stream into an audio voice signal comprising the steps of:

(a) buffering a received parametric voice data stream having a plurality of pitch periods [and loading said buffered frame data into a buffer];

(b) constructing an estimated spectrum of excitation within each pitch period by breaking down the frequency spectrum into regions based on a cutoff frequency, wherein said construction comprises the steps of:

(i) computing a Fourier magnitude for each region, wherein the resultant computed Fourier [magnitudes] magnitude for at least one of said regions is then scaled by a gain factor computed for that region,

(ii) computing phase within each region, wherein the resultant phase for at least one of said regions has been modified by use of a weighted random phase, and

(iii) converting said Fourier magnitude and said phase within each region to a time domain representation by the computation of an inverse discrete Fourier transform; and

(c) generating an analog voice signal from said time domain representation.

21. (Once Amended) A method as recited in claim 20, wherein said regions into [through] which the frequency spectrum is broken down [into] comprise:

(a) a lower region wherein Fourier magnitudes directly determine the spectrum;

(b) a transition region wherein Fourier magnitudes are scaled down by a linearly decreasing weighting factor that drops from unity to a nonzero positive value dependent on the cutoff frequency of the current frame; and

(c) an upper region wherein Fourier magnitudes are scaled down by a weighting factor dependent on the cutoff frequency of the current frame.

23. (Once Amended) A down-transcoder apparatus which receives an encoded frame-based voice data stream and converts it into a superframe-based encoded voice data stream, comprising:

(a) a superframe buffer for collecting a number of frames of parametric voice data and extracting bits representing frame-based voice parameters;

(b) a decoder for inverse quantizing the bits for each frame of [parameter] parametric voice data into quantized parameter values for each frame; and

(c) a superframe encoder for collecting said quantized frame-based parameters for the group of frames within the superframe, producing a set of parametric voice data, and quantizing and encoding said parametric voice data into an outgoing digital bit stream.